AMENDMENTS TO THE CLAIMS:

This listing of claims will replace all prior versions, and listings, of claims in the application:

1.-19. (Cancelled)

20. (Previously Presented) A method of transmitting a recording comprising a sequence of data packets , the method comprising:

a server commencing transmission of the recording over the network to a receiver;

the receiver holding received data in a receiver buffer; and

at the server, a control unit:

analyzing the entire said sequence of data packets to determine where a point in the transmission of the recording is reached at which, if the receiver were to commence decoding data already transmitted and held in said receiver buffer, said receiver buffer would not underflow;

continuing transmission to the receiver,

wherein, said analyzing comprises analyzing the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest first section that meets the condition that it covers a playing time interval greater than or equal to its respective maximum timing error value; and

causing the receiver to commencing playing of received data only after said first section has been received.

21. (Previously Presented) A method according to claim 20, wherein after transmission of said first section, said receiver is caused to play by transmitting an instruction to the receiver to commence playing.

22. (Previously Presented) A method according to claim 20, comprising:

transmitting to the receiver an instruction specifying the first section and wherein the receiver is caused to commence playing when said receiver recognizes that the first section is in said receiver buffer.

23. (Previously Presented) A method according to claim 20, in which:

at the server, said analyzing comprises computing maximum timing error values for different sections of the sequence, and

at the receiver, the maximum timing error values are compared with the buffer contents to recognize when said first section is in the buffer.

24. (Previously Presented) A method according to claim 20, comprising:

withholding transmission of an initial part of the recording until a remainder of said first section has been transmitted; and

transmitting said initial part;

wherein the receiver commences playing only when said initial part is received.

25. (Previously Presented) A method according to claim 20, including:

performing the analysis in advance of said transmission of said recording to the receiver; and

marking an identified section in the recording prior to transmission of said identified section.

26. (Previously Presented) A method according to claim 20, wherein said analyzing includes:

computing, in advance, timing error values corresponding to a plurality of transmitting data rates and storing the timing error values; and

subsequently estimating therefrom an error value corresponding to an actual transmitting data rate.

27. (Previously Presented) A method according to claim 20, in which the analyzing comprises:

testing a timing error parameter evaluated for successive portions of the recording,

wherein the timing error parameter is first calculated in respect of a first or early portion of the recording and the timing error parameter for subsequent portions is obtained by updating the parameter obtained for the preceding portion .

28. (Previously Presented) A method according to claim 20, in which the recording is a video recording.

29. (Previously Presented) A method according to claim 20, in which the recording is an audio recording.

30. (Previously Presented) A method according to claim 20, wherein:

the recording is transmitted in a network from the server to the receiver at a fluctuating transmitting data rate which is not known when the recording in its entirety is analyzed to identify a first section at the beginning of the recording which meets a condition that it covers a playing time interval greater than or equal to a maximum of an extent to which a transmission time of a respective following section exceeds its playing time interval for a following data section of any length; and wherein

said causing the receiver comprises a control message which causes the receiver to commence playing of received data only after said first section has been received.

31. (Previously Presented) A method according to claim 20, wherein the recording is transmitted over a network and is to be played in real time by the receiver.

32. (Cancelled)

33. (Previously Presented) A method according to claim 20,

wherein a timing error parameter is calculated as the extent to which the transmission time of a following section of the recording, said following section being a section following said first section, exceeds the playing time interval for said following section, and wherein the

maximum timing error value is the maximum of the timing error parameters over the set of all possible following sections of any length from said first section.

34. (Previously Presented) A method according to claim 20, wherein

a timing error parameter is calculated as the extent to which the transmission time of a following section of the recording, said following section being a section following said first section, exceeds the playing time interval for said following section, and wherein the maximum timing error value is the maximum of the timing error parameters over the set of all possible following sections of any length from said first section,

wherein said timing error parameter is calculated as the sum of the timing errors from the end of said first section to the end of the following section and a timing error is the difference between the transmission time of a portion and the playing time of a portion.

35. (Currently Amended) An apparatus arranged to transmit a recording comprising a sequence of data packets stored in a data store accessible by a server comprising a control unit and a transmitter over a network to a receiver comprising a receiver buffer, the apparatus comprising:

said server comprising a control unit and a transmitter,

said receiver comprising a receiver buffer, wherein

when the server commences transmission of the recording over the network to the receiver, ~~;~~

the receiver buffer is arranged to hold received data ~~in said receiver buffer~~until the receiver is caused to commence playing of received data, ~~;~~

wherein ~~until~~ at the server, the recording is analyzed by said control unit~~: has~~ analyzing~~ed~~ the entire said sequence of data packets to determine where a point in the transmission of the recording is reached at which, if the receiver were to commence decoding data already transmitted and held in said receiver buffer, said receiver buffer would not underflow;

wherein, the entire said sequence of data packets is analyzed to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said plurality of first sections that meets the condition that it covers a playing time interval greater than or equal to its respective maximum timing error value, and wherein

the receiver is caused to commence playing of received data after the shortest said first section that meets the condition has been received.

36. (Previously Presented) An apparatus according to claim 35,

wherein the timing error parameter which calculates the extent to which the transmission time of a section of the recording following said first section exceeds the playing time interval for said first section is calculated as the sum of the timing errors from the end of said first section to the end of the following section; and

wherein the maximum timing error value is the maximum of the timing error parameters over the set of all possible following sections of any length from said first section.

37. (New) A method of transmitting a recording comprising a sequence of data packets, the method comprising:

commencing, using a server, transmission of the recording over the network to a receiver configured to hold received data in a receiver buffer; and

at the server, a control unit:

analyzing the entire said sequence of data packets to determine where a point in the transmission of the recording is reached at which, if the receiver were to commence decoding data already transmitted and held in said receiver buffer, said receiver buffer would not underflow;

continuing transmission to the receiver, wherein, said analyzing comprises analyzing the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said first sections that meets the condition that it covers a playing time interval greater than or equal to its respective calculated maximum timing error value; and

causing the receiver to commence playing of received data only after said shortest first section that meets the condition has been received.

38. (New) A method of transmitting a recording comprising a sequence of data packets over a network to a receiver, the method comprising:

analyzing, prior to the transmission of the entire recording, the entire sequence of data packets of the recording to determine where a point in said transmission of the recording is

reached at which, if the receiver were to commence decoding data already transmitted over said network by said server and held in a receiver buffer of said receiver, said receiver buffer would not underflow,

wherein said analyzing comprises analyzing the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said first sections that meets the condition that it covers a playing time interval greater than or equal to its respective calculated maximum timing error value;

transmitting analyzed data packets over the network to said receiver arranged to hold received data in a receiver buffer; and

causing the receiver to commence playing of received data only after said shortest first section that meets the condition has been received.

39. (New) A method of transmitting a recording comprising a sequence of data packets over a network to a receiver, the method comprising:

analyzing, using a control unit, the entire said sequence of data packets to determine where a point in a transmission of the recording is reached at which, if the receiver were to commence decoding data already transmitted over said network and held in a receiver buffer, said receiver buffer would not underflow, wherein said analyzing comprises analyzing the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the

transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said first sections that meets the condition that it covers a playing time interval greater than or equal to its respective calculated maximum timing error value;

transmitting, using a transmitter, analyzed data packets over the network to a receiver arranged to hold received data in a receiver buffer; and

causing the receiver to commence playing of received data only after said shortest first section that meets the condition has been received.

40. (New) An apparatus comprising:

a transmitter arranged to transmit a recording comprising a sequence of data packets stored in a data store accessible to the server to a receiver comprising a receiver buffer; and

a control unit arranged to analyze said data packets prior to their transmission by said transmitter,

wherein the control unit is arranged to analyze the entire said sequence of data packets of said recording to determine where a point in the transmission of the recording is reached at which, if the receiver were to commence decoding data packets from said sequence already transmitted and held in said receiver buffer, said receiver buffer would not underflow;

wherein, the entire said sequence of data packets is analyzed to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said plurality of first sections that meets the

condition that it covers a playing time interval greater than or equal to its respective maximum timing error value, and

means to cause, after the transmitter has transmitted said first section to the receiver, said receiver to commence playing of received data after said first section has been received.

41. (New) A method of transmitting a recording comprising a sequence of data packets, the method comprising:

commencing, using a server, transmission of the recording over the network to a receiver; and

causing the receiver to commence playing of received data only after said shortest first section that meets a pre-determined condition has been received;

wherein said pre-determined condition was determined from an analysis of the entire said sequence of data packets to determine where a point in a transmission of the recording would be reached for predetermined transmission conditions at which, if the receiver were to commence decoding data already transmitted under said predetermined transmission conditions and held in said receiver buffer, said receiver buffer would not underflow, the analysis comprising an analysis of the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time under said predetermined timing conditions of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest first section that meets the condition that it covers a playing time interval greater than or equal to its respective maximum timing error value.

42. (New) A method of analyzing a recording comprising a sequence of data packets for transmission over a network to a receiver, the analysis determining when the receiver is to start to play received data packets, the method comprising:

analyzing, prior to the transmission of the entire recording, the entire sequence of data packets of the recording to determine where a point in said transmission of the recording is reached at which, if the receiver were to commence decoding data already transmitted over said network by said server and held in a receiver buffer of said receiver, said receiver buffer would not underflow,

wherein said analyzing comprises analyzing the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said first sections that meets the condition that it covers a playing time interval greater than or equal to its respective calculated maximum timing error value, and

wherein, when the analyzed data packets are transmitted over the network to said receiver arranged to hold received data in said receiver buffer, the receiver is caused to commence playing of received data only after said shortest first section that meets the condition has been received by the receiver.

43. (New) An apparatus arranged to analyze a recording comprising a sequence of data packets for transmission over a network to a receiver, the analysis determining when the receiver is to start to play received data packets, the apparatus comprising:

a control unit arranged to analyze, prior to the transmission of the entire recording, the entire sequence of data packets of the recording to determine where a point in said transmission of the recording is reached at which, if the receiver were to commence decoding data already transmitted over said network by said server and held in a receiver buffer of said receiver, said receiver buffer would not underflow,

wherein said control unit analyzes the entire said sequence of data packets to calculate for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said first sections that meets the condition that it covers a playing time interval greater than or equal to its respective calculated maximum timing error value,

whereby, when the analyzed data packets are transmitted over the network to said receiver arranged to hold received data in said receiver buffer, the transmission includes data generated by said analysis arranged to cause said receiver to commence playing of received data only after said shortest first section that meets the condition has been received by the receiver.

44. (New) A non-transitory data store of analyzed data recordings, the data store being accessible by a transmitter arranged to transmit a said analyzed data recording accessed from said data store to a receiver comprising a receiver buffer for holding data before the receiver commences playing data, wherein a stored analyzed data recording comprises:

a sequence of data packets analyzed by a control unit, the analysis comprising an analysis of the entire said sequence of data packets to determine when, in a transmission of the recording at a pre-determined rate, a point in the recording is reached at which, if the receiver were to commence decoding data already transmitted at said pre-determined rate and held in said receiver buffer, said receiver buffer would not underflow;

wherein the analysis of the entire said sequence of data packets of a recording calculates for each of a plurality of first sections in the recording a maximum timing error value calculated as the maximum of the extent to which the transmission time of the respective following section of the recording exceeds its playing time interval for a following section of any length, wherein said point is determined as the end of the shortest of said plurality of first sections that meets the condition that it covers a playing time interval greater than or equal to its respective maximum timing error value,

wherein the analysis provides control data to be stored with said data recording, the control data causing the receiver to commence playing of received data after the shortest said first section that meets the condition has been received.